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EXAMINER

HAN, QI

ART UNIT

PAPER NUMBER

2654

DATE MAILED: 03/10/2006

Please find below and/or attached an Office communication concerning this application or proceeding.



### **DETAILED ACTION**

1. The text of those sections of Title 35, U.S. Code not included in this action can be found in a prior Office action.

#### ***Response to Amendment***

2. This communication is responsive to the applicant's amendment dated 12/09/2005. Applicant amended claims 1, 7-8, 17, 23-24, 28-29, 31 and 37, cancelled claims 40, 42, 46 and 48, and added new claims 49-53 (see the amendment pages 2-14).

The examiner withdraws the claim rejection under 35 USC 112 2<sup>nd</sup>, because the applicant cancelled claims (see the amendment pages 12-13).

The examiner withdraws the claim rejection under 35 USC 103 regarding claims 1, 7-8, 14-17, 23-24, 28-29, 31, 37, 39, 41, 43, 45, 47 and 49-53, because the applicant amended claims (see the amendment pages 2-14).

#### ***Claim Rejections - 35 USC § 103***

3. Claim 38 is rejected under 35 U.S.C. 103(a) as being unpatentable over Pickett (US 2002/0001302 A1), in view of Sharman et al. (US 6,100,882), hereinafter referenced as Sharman.

Regarding **claim 38**, Pickett discloses systems and methods for multiple mode voice and data communications using intelligently bridged TDM and packet buses and methods for performing telephony and data functions using the same (title), in which VoIP communications attempts to provide reasonable voice communications over data/packet networks by allowing voice and signaling information to be transported over the data/packet network, and an IP network typically is used to transport the calls, which generally may be over an intranet or over the Internet (paragraph 367) that inherently provides packet based communication session for voice and text data, which corresponds to the claimed “communicating voice and text associated with a packet based voice communications session”. Pickett further discloses that:

“receiving voice information from a local participant in a packet-based voice communications session having at least one remote participant”, (paragraph 194 and Figs. 3 and 13C, ‘computer 24 (local and/or remote participants) is coupled to communications system 50 (network) over packet bus 80A’, ‘a microphone (for receiving voice information)’, ‘through an appropriate packet standard’, ‘H.323’ ‘for transmission to a remote computer’; paragraph 370, ‘H.323 terminal... used for real-time bi-directional multimedia communication’);

“detecting a degradation in a quality of the packet-based voice communications session”; (paragraph 105, ‘line quality assessment (interpreted as detecting)’ ‘capability enables ... to link status indicators’, ‘the line condition... (e.g. “speed grading” or “speed characterization” of individual lines) can be measured (detecting a degradation)’ ; paragraph 363, ‘enhance voice quality’, ‘dynamically adjustable jitter buffer, packet-loss correction, and noise-level matching’, which suggests detecting a degradation during the voice communication);

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“converting the [local] voice information into [local] text”, (paragraph 297, ‘speech/voice recognition’, ‘speech to text conversion, compression, translation’);

“generating a first stream of packets encoding the text” (paragraph 297, ‘speech to text conversion’, ‘compression (broadly interpreted as encoding)’, ‘data stream from the LAN, WAN ...may be desirably coupled to resources’ that inherently includes packet based transmission; paragraph 298, ‘processes the voice data stream into another form (e.g. email (text))’);

“generating a second stream of packets encoding the voice information” (paragraph 74, ‘coding/decoding function’, ‘voice compression’, ‘voice communication using an Internet protocol (“IP”) or other voice over other network protocol’ that inherently uses packet-based transmission);

“communicating the first stream of packets using transmission control protocol (TCP)”, (paragraph 194, ‘processes the packetized data stream... in a suitable form/protocol (such as TCP/IP) for transmission to a remote computer’; paragraph 297, ‘speech to text conversion (corresponding to the first stream)... thus data stream from the LAN, WAN, modem (through communicating)... ’, which suggest that text type of data stream can be communicated in the LAN that is packet-based network; paragraph 298, ‘another form (including text form) may be stored, send (communicate) over the WAN or LAN’);

“communicating the second stream of packets using user datagram protocol (UDP)” (paragraphs 374 and 388, ‘addressing in VoIP (the second stream)’, ‘UDP header containing source and destination sockets’, ‘voice data is traveling over a data network inside TCP or UCP packets’);

“receiving packets encoding remote voice information and remote text from the remote participant”, (Pickett: paragraph 194 and Fig. 3, ‘computer terminal (also H.323 terminal) 24’ (one of the terminal 24 acts as a remote terminal during communication); paragraph 370, ‘multimedia communication application(s)’, ‘H.323 terminal... used for real-time bi-directional multimedia communication’);

“outputting the remote voice information using an acoustic output device”, (paragraph 194 and Fig. 3 and 13C, ‘computer 24 (Fig. 13C) includes ... speaker’; Sharman: Fig. 1, blocks 10 and 20, ‘computer workstation’ and ‘telephone’; paragraph 370, ‘multimedia communication application(s)’, ‘H.323 terminal... used for real-time bi-directional multimedia communication’);

“in response to detecting the degradation in the quality of the packet-based voice communications session, displaying the remote text using a visual output device”, (paragraph 105, ‘line quality assessment (detecting)’, ‘link status information’, ‘lines condition’ (inherently including degradation) ‘can be incorporated into a visual representation of the system’ and ‘easily viewable remotely...’; paragraph 194 and Fig. 3, ‘computer terminal (also H.323 terminal) 24’; paragraph 72, ‘processor/system resources 70 also may include a display device’); Sharman: Fig. 9, blocks 945 and 955).

In addition, Pickett discloses that the system provides Voice over IP (VoIP) technique (paragraph 361), uses H.323 standard (paragraph 368), and uses H.323 terminals that can either be a PC or a standalone device and provides audio communications while optionally supporting video or data communications (paragraph 361), which further supports to implement the functionality as stated above because both VoIP and H.323 are packet-based communications and H.323 supports multimedia communications including audio and text.

Even though Pickett discloses the capability of generating and communicating packeted text data stream and voice data stream and centralized speech-to-text conversion, as stated above, Pickett does not expressly teach “determining that the packet-based voice communication session provides for a text communication session”. However, this feature is well known in the art as evidenced by Sharman who discloses textual recoding of contributions to audio conference using speech recognition (title), comprising a distributed system performing speech recognition to convert speech to text at local workstation (including converting the local speech into a text locally) (column 2, lines 50-65) that can be used for audio conference (abstract), including ‘transmitting the local speech input (voice communication session) to the other participant (remote)’ ‘plus the corresponding text (text communication session) equivalent transmitted’ (col. 2, line 30-66). Therefore, it would have been obvious to one of ordinary skill in the art at the time the invention was made to modify Pickett by specifically providing speech transmission plus the corresponding text transmission, as taught by Sharman, for the purpose of offering voice and text with natural conversation and providing optional feature of automatic translation for multilingual conferences (Sharman: column 3, lines 10-20).

***Allowable Subject Matter***

4. Claims 1, 7-8, 14-17, 23-24, 28-29, 31, 37,39, 41, 43, 45, 47 and 49-53 are allowed.

The following is an examiner’s statement of reasons for allowance:

Regarding independent **claims 1, 8, 17, 24 and 31**, the instant application is directed to a method, interface, telephony communication software, device, and communication system for communicating voice and text associated with a packet-based communications session. Each

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independent claim, combine other well-known feature in the art, identifies the uniquely distinct features of: converting a voice information into local text; generating a first stream of packets encoding the local text; determining a degradation in a quality of the packet-based voice communication; in response to detecting the degradation in the quality of the packet-based voice communications session, communicating the first stream (encoded and converted text of the voice information) of packets to the remote participant using transmission control protocol (TCP); and communicating the second stream (encoded voice information) of packets to the remote participant using user datagram protocol (UDP).

The prior art of record, Pickett (US 2002/0001302 A1), Sharman et al. (US 6,100,882), and White et al. (US 6,408, 272 B1 Pickett (US 2002/0001302 A1), provided numerous techniques and approaches of packet-based voice/data communication, including using voice over Internet protocol with H.320 and H.323 data streams, providing speech recognition and conversion for the voice data and certain quality/degradation detection, using TCP or UDP protocol for various packet communications, including application of audio conference with recognized text transmission, and providing distributed voice user interface and display. However, the combined features stated above, are not anticipated by, nor made obvious over the prior art of the record.

5. Any comments considered necessary by applicant must be submitted no later than the payment of the issue fee and, to avoid processing delays, should preferably accompany the issue



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fee. Such submissions should be clearly labeled "Comments on Statement of Reasons for Allowance."

***Conclusion***

6. **THIS ACTION IS MADE FINAL.** Applicant is reminded of the extension of time policy as set forth in 37 CFR 1.136(a).

A shortened statutory period for reply to this final action is set to expire THREE MONTHS from the mailing date of this action. In the event a first reply is filed within TWO MONTHS of the mailing date of this final action and the advisory action is not mailed until after the end of the THREE-MONTH shortened statutory period, then the shortened statutory period will expire on the date the advisory action is mailed, and any extension fee pursuant to 37 CFR 1.136(a) will be calculated from the mailing date of the advisory action. In no event, however, will the statutory period for reply expire later than SIX MONTHS from the mailing date of this final action.

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Any inquiry concerning this communication or earlier communications from the examiner should be directed to Qi Han whose telephone numbers is (571) 272-7604. The examiner can normally be reached on Monday through Thursday from 9:00 a.m. to 7:00 p.m. If attempts to reach the examiner by telephone are unsuccessful, the examiner's supervisor, Richemond Dorvil, can be reached on (571) 272-7602.

Information regarding the status of an application may be obtained from the Patent Application Information Retrieval (PAIR) system. Inquiries regarding the status of submissions relating to an application or questions on the Private PAIR system should be directed to the Electronic Business Center (EBC) at 866-217-9197 (toll-free) or 703-305-3028 between the hours of 6 a.m. and midnight Monday through Friday EST, or by e-mail at: [ebc@uspto.gov](mailto:ebc@uspto.gov). For general information about the PAIR system, see <http://pair-direct.uspto.gov>.

QH/qh  
March 2, 2006

  
RICHEMOND DORVIL  
SUPERVISOR EXAMINER